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PERFORMANCE OF TOKEN PASSING LAN'S FOR INTEGRATED TRANSMISSION OF VOICE AND DATA

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قياس كفاءة شبة حاسب محليـــه مستخدمـــه فى نقل الصوت والمعلومـــات

خلاصة :

فى شبكات الحاسب الحديثة يمكن نقل معلومات الصوت الرقمية جنبا إلى جنـــب مـــع البيانات. و لكي نتمكن من ذلك يجب اختيار البرتوكول المناسب للتعـــامل مـــع البيانـــات و الصوت دون أن يحدث تأخير لمعلومات الصوت و هذا البحث يقدم طريقة مستحدثة لحســـاب أفضل أداء لشبكة حاسوب محلية تتعامل بيانات المعلومات و الصوت. وقد استخدم الخـوارزم الجيني فى حساب لأفضل أداء للشبكة و ذلك باختيار لأفضل قيم لمتوسط البيانات

Abstract

The performance of integrated transmission of voice and data on token passing ring LAN's is investigated in this paper. The aim is to optimize the performance of the integrated transmission of voice and data on token passing ring LAN's by optimizing the average data packet size. Genetic algorithm is used to solve the optimization problem.

Introduction

Local area networks (LAN's) have conceived as low cost high throughput structure for data traffic transport. Interest in LAN's has been growing in the last few years in view of integrating real-time services, and voice is the first natural candidates among real-time sources to be considered for a service integration. The round-robin nature of token ring networks which provides reasonable packet delays at light loads ,and allows the packet delay to be bounded at high loads makes them very attractive as integrated service LAN's [1].

Studying the performance of these networks is an important issue to obtain the best use of it. But the fact that performance objectives vary tremendously from one type of services to another, presents a real challenge to designers of network algorithms and protocols [2]. For example, for data traffic, real-time data delivery is not of primary

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importance, but strict error control and recovery procedures are required
On the other hand, for voice traffic, excessive delays can have seric
disruptive effects on human conversation [3].

Many protocols and algorithms are introduced to perform the best performance of these networks. An access protocol for efficient voice/data integration in a token ring network which is based on the adoption of a variable size of voice packet that is determined by the actual load conditions is introduced in [1] which consider voice source to be always in talkspurt state. Another protocol which models voice as having alternating talkspurts and silences, with generation of voice packet at a constant rate during talkspurts and no packet generation during silence periods is introduced in [3]. The efficiency of such networks greatly increased through the use of sophisticated scheduling and dropping algorithms within the queues formed at the network access points and use cost-based scheduling and cost-based dropping to optimize the network performance [2].

In this paper, the network performance is optimized through the optimization of the data packet size. The genetic algorithm is used to achieve the optimum performance along the network model reported in [1].

The organization of this paper is as follows:

Network and traffic model is introduced in section (1). In section (2) the access protocol is considered. In section (3) the basic features of the access protocol is discussed. In section (4) the performance evaluation of the network is introduced. The genetic algorithm is introduced in section (5). Seetion (6) shows the simulation results, and section (7) contains conelusions and discussions.

I. Network and traffic model

The following notations are introduced to describe the network model and the traffic process [1].

N : number of stations evenly spaced on the ring each supporting either a voice or a data source.

d : ring length (Km).

C_r : network bit rate (bits/sec).

τ : signal propagation delay (sec/Km).

b_s : *signal latency in a station (bit).*

N_d : number of data sources, presumed to be poissonian.

g : *average traffic ordered by a data source (packet/sec).*

l_d : data packet length, with average value L_d (bit).

N_v : *number of voice sources.*

C_v : speech coding rate (bit/sec).

l_v :voice packet length, with average value L_v (bit).

L_o : *total number of framing overhead bits.*

L_{he} : number of header bits in the frame format.

II. Access Protocol

The access protocol described is based on two concepts. First, voice information is transferred by means of variable size packets and second, a suitable scheme is adopted to give voice packets priority over data packets that guarantees a minimum data bandwidth fairly allocated among stations [1].

III. Basic Features of the Access Protocol

The characteristics of the access protocol includes three features [1].

1- Voice packets have a variable size and the data packets transferred through the network cannot be longer than L_{dmax} bits (users information units longer than L_{dmax} are segmented into more than one data packet to satisfy this bound).

This feature implies that, the voice packet size l_s depends on the voice cycle period, that is the time between two consecutive events of access permission receipt by a voice station. The limit L_{dmax} on the data packet size, together with the access mechanism itself, allows the voice cycle period to be bounded. The adoption of a variable size of voice packets corresponds to maximizing the network throughput and minimizing the access time of voice packets. The voice packet size increases with the number N_v of voice source, owing to the longer cycle period caused by a larger number of voice sources. With a very small number of voice stations, voice packets become very small. Bandwidth is wasted as the overhead per packet becomes too large, and no real need exists for the smallest voice packet delays, e.g. less than 1ms. For this reason, a lower bound L_{vmin} is defined for the size of voice packets, meaning that a voice packet becomes ready to be transmitted only when the coder has generated L_{vmin} bits. This causes a decrease in the average voice cycle period for a small number of voice users, and hence an increase in the data bandwidth for low voice traffic levels. Since a large L_{vmin} determines a better performance of data traffic, L_{vmin} should be selected as the largest value corresponding to a reasonable minimum end to end delay of voice packets. This minimum delay is essentially given by the packetization delay L_{vmin}/C_v. Note that the delay requirement of voice packets can vary from source to source. For example, a local call within the LAN can well accept packet delays of the order of 200ms, for the packet voice transfer in the LAN.

2- Voice stations access the network with priority over data stations, by issuing a request to receive the access right, and the delay between the access request and the subsequent voice packet transmission is suitably bounded.

3- A certain minimum data bandwidth is always guaranteed, which is fairly allocated among all data stations.

The second and third features imply that, two kinds of cycles are defined, a voice cycle and a data cycle, each allowing only one kind of stations to transmit. Two occurrences of a cycle type are always interleaved by an occurrence of the other cycle type and a voice (data)cycle begins at the start of transmission of the first voice (data) token/frame in the cycle. Every voice station is given the access opportunity in a voice cycle and at least one data station can issue an information frame in a data cycle. The data-to-voice cycle transition occurs either upon a request by a voice station or after a token round without any data frame transmitted. In this protocol a voice station can transmit only during a voice cycle and can request the network access only when a data frame is actually relayed by the station. (The term frame relayed by a station stands for a frame transmitted but not generated by the station). The voice-to-data cycle transition takes place when all voice stations have been given the access opportunity.

The above protocol corresponds to provide network access to voice stations on regular bases and to guarantee at the same time a given data bandwidth.

IV. Performance Evaluation

To evaluate the performance of the single token access protocol the following assumptions are considered:

$l_{min}=0$, data packets have a fixed length L_d, voice silences are not suppressed (i.e. each voice source generates C_v bits/sec, as the source is always presumed to be in the talkspurt state), the queues of data stations are never empty, and at least one voice source offers traffic to the network.

In the network under study, the ring latency b_r in bit transmission time is given by:

$$b_r = (N_v + N_d)b_s + d\tau C_r \qquad (1)$$

Depending on the voice and the data packet sizes, a time gap can separate the transmission of an information frame and the following token. This occurs if the frame transmission is completed before the frame header is completely received by the same source station, that is

$$l_d + L_o \leq b_r + L_{he} \qquad (2\text{-a})$$

$$l_v + L_o \leq b_r + L_{he} \qquad (2\text{-b})$$

Without loss of generality, steady-state conditions are considered in the network, so that (2-a) is either always or never satisfied for a given number N_v of voice sources. Therefor, those limiting cases in which the access mechanism detehmines voice packet sizes both below and above the threshold of eqn.(2) are not considered here.

Let t_{td} and t_{tv} denote the transmission times of a data frame and a voice frame, respectively, and t_{cv} indicates the voice cycle period, defined as the time elapsed between two consecutive receipts of the voice token by the same voice station. Let T_{td}, T_{tv} and T_{cv} be the average values of t_{td}, t_{tv} and t_{cv} respectively, hence by considering that the voice packet size is

proportional to the average voice cycle period, yields

$$L_v = C_v T_{cv} \qquad (3)$$

and

$$T_{tv} = \frac{L_v + L_0}{C_r} \tag{4}$$

$$T_{td} = \frac{L_d + L_0}{C_r} \tag{5}$$

Applying eqn.(3) into eqn.(4), then

$$T_{tv} = \frac{C_r T_{cv} + L_0}{C_r} \tag{6}$$

As each voice station issues a voice frame between any two data frame transmissions, T_{cv} also represents the data frame period, that is the average time between two consecutive starts of data cycle. In a data frame period, each active voice station generates a voice frame, one data station issues a data frame, and a certain propagation time is required for the token to move between adjacent stations. (two data [voice] stations "a" and "b" are said to be adjacent if "b" is the first data [voice] station to receive a data [voice] token issued by "a", or vice versa).

The average token propagation time in a data frame period is $b_r(1+1/N_d)$ bits. In fact, it takes an average time of b_r/N_d bits for the data token transfer between two adjacent data stations and b_r bits for a ring round of the voice token. Thus, by taking eqns. (2) into consideration, the average voice cycle period can be written as:

$$T_{cv} = \frac{\max(L_d + L_0, b_r + L_{he}) + N_v \max(L_v + L_0, b_r + L_{he}) + b_r(1 + 1/N_d)}{C_r} \tag{7}$$

Taking into account that no values of N_v satisfies eqn. (2), the average voice cycle period can be reexpressed as [5]:

$$T_{cr} = \frac{\max(L_d + L_0, b_r + L_{he}) + N_v(L_v + L_0) + b_r(1 + 1/N_d)}{C_r} \tag{8}$$

Applying eqn.(3) into eqn.(8), then

$$T_{cv} = \frac{\max(L_d + L_0, b_r + L_{he}) + N_v(C_v T_{cv} + L_0) + b_r(1 + 1/N_d)}{C_r} \tag{9}$$

By some algebraic manipulation, it can easily be shown that:

$$T_{cr} = \frac{\max(L_d + L_0, b_r + L_{he}) + N_v L_0 + b_r(1 + 1/N_d)}{(C_r - N_v C_v)} \tag{10}$$

In the study reported here, three parameters are considered in the performance evaluation. First, The maximum net work throughput ρ_{max}, which is defined as the maximum ratio between the average traffic on the network, i.e. the average bit rate resulting from frame transmissions, and the network capacity C_r. It can be formulated mathematically by:

$$\rho_{max} = \frac{L_d + L_0 + N_v(L_v + L_0)}{C_r T_{cv}} \tag{11}$$

Second, the normalized average data bandwidth expressed as:

$$W_d = \frac{T_{td}}{T_{cv}} \tag{12}$$

Applying eqn.(5) and (10) into eqn.(12),then

$$W_d = \frac{(L_d + L_0)(1 - N_v C_v / C_r)}{\max(L_d + L_0, b_r + L_{he}) + N_v L_0 + b_r(1 + 1/N_d)} \tag{13}$$

Third, the average voice packet end - to - end delay T_{cv}. It is one of the most constraining characteristics for voice transmission in the necessity to maintain little or no delay, while this delay is not important in data transmission [5]. Therefore, the average voice packet end-to-end delay T_{ev} has an important meaning for such a study. By assuming that the distance between source and destination stations to be half ring, then the average voice packet end-to-end delay can be formulated as:

$$T_{ev} = T_{cr} + T_{tv} + \frac{b_r}{2C_r} \tag{14}$$

applying eqn. (6) into eqn. (14), then

$$T_{ev} = T_{cv}(1 + \frac{C_v}{C_r}) + \frac{L_0 + b_r/2}{C_r} \tag{15}$$

V. The Genetic Algorithm

Genetic algorithm combines the survival of the fittest with the innovative flair of human search. It is considered as a form of the random search method which is suitable for discontinuous and multimodal problems. The genetic algorithm adopted here [4] can be summarized as:

Step 1 : [Initialization]

n=0

Heuristically choose a population of m candidates

$\{x_1^n, x_2^n, ..., x_m^n\}$

$x_{op}(optimal \quad candidates) = x_1^0$

Step 2 : [evaluate fitness]

Do I =l, m

Find fitness $f(x_1^n)$ where $f(.)$ is a defined objective

function.

If $F(x_i^n) > f(x_{op})$

$x_{op} = x_i^n$

END IF

END DO

Step 3 : [check the termination criterion]

If the termination criterion is satisfied

GO to step 5

step 4 : [produce the next generation]

create $\{\hat{x}_1^{n+1}, \hat{x}_2^{n+1}, \ldots, \hat{x}_m^{n+1}\}$ from $\{x_1^n, x_2^n, \ldots, x_m^n\}$

according to the weights of fitness $f(x_i^n)$, i=1,2,....m

Generate $\{x_1^{n+1}, x_2^{n+1}, \ldots, x_m^{n+1}\}$ by crossover and mutation

n = n + 1

Go To Step 2

Step 5 : [END]

PRINT the optimal candidate x_{op} and its fitness $f(x_{op})$

In this paper, The objective function for a given candidate x_i^n in the nth generation is defined as:

$$f(x_i^n) = \left(W_d + \frac{200 - T_{ev}}{200 - 1} \right) / 2$$

VI. Simulation Results

From the preceding discussion, it is found that the network performance can be optimized by optimizing the L_d. The genetic algorithm is used to compute the optimum value of L_d, hence the optimum value of L_v is then computed using eqn.(3).

In the simulation experiments, the following parameters are chosen for the network: b_s=2bits, $\tau=5\times10^{-6}$sec/Km, L_0=168bits, C_v=64Kbps, L_{hc}=120bits, C_r=1, 4 and 16Mbps, and d =500m. These parameters are typical values for a coaxial cable LAN adopting PCM voice coding scheme [3] and framing overhead specified in the IEEE 802.5 standard for token ring [1].

Figures (1), (2) and (3) show the relationship between N_v and $L_{doptimum}$ (for different values of N_d (number of data stations) and C_r = 1, 4 and 16Mb/sec respectively. It can be seen that as N_v increases, $L_{doptimum}$ increases until a specific N_v, this is because increasing N_v increases the average voice cycle period and so, the amount of data transferred decreases so $L_{doptimum}$ increases to compensate the decrease in data transferred. After this specific N_v the value of $L_{doptimum}$ optimum average data packet size) decreases with more increase of N_v. This is because with the increase of N_v the average voice packet end-to-end delay (T_{ev}) increases and so $L_{dopimum}$ decreases to maintain T_{ev} within its maximum value.

Figures (4), (5) and (6) show the relationship between ρ_v (the voice load,$\rho_v=N_vC_v/C_r$) and ρ_{max}(maximum network throughput) for different values of L_d and $L_{doptimum}$ for C_r=1,4 and 16Mb/sec respectively. From these figures, it can be seen that ρ_{max} significantly increases with the increase of L_d for small voice loads but, with more increase in the voice load the maximum network throughput becomes dependant on L_d. This is because with small voice loads the network is not fully occupied so there is a free space which if it is allocated to the data users (by increasing L_d) ρ_{max} will increase. But, with more increase in voice loads the network will be highly occupied with the voice users (due to the priority of voice users over data users) so, ρ_{max} will not be significantly affected with the increase of L_d.

Figures (7), (8) and (9) show the relationship between N_v and W_d for different values of L_d and for $L_{doptimum}$ for C_r=1, 4 and 16Mb/sec respectively. It can bee seen that W_d significantly increases with L_d, unless large values of N_v. is used. On the contrary, W_d decreases as N_v increases by approaching the value 0 when the voice throughput saturates the network capacity

Figures (10), (11) and (12) show the relationship between N_v and T_{cv} for different values of L_d and for $L_{doptimum}$ for C_r=1, 4 and 16Mb/sec respectively. These figures show that the data packet size significantly affects the average packet delay only for small number of voice stations. An increasing number of voice stations makes the voice cycle period less dependant on the data frame size, as only one data station transmits after the transmission by N_v voice stations. Also, these figures show that, although, minimum T_{cv} is not obtained for the curves of $L_{doptimum}$ but, the values of T_{cv} for the curve that uses $L_{doptimum}$ still within the limits of T_{cvmin} and T_{cvmax} that specified before. In the same time the maximum values of ρ_{max} and W_d are kept.

From the above discussion, it can be concluded that, optimum performance is obtained for C_r=4 and 16 Mb/sec. For C_r=1 Mb/sec the optimization process does not provide significant improvement to the network performance. Moreover, the maximum number of voice users on the network change with the change of the bit rate.

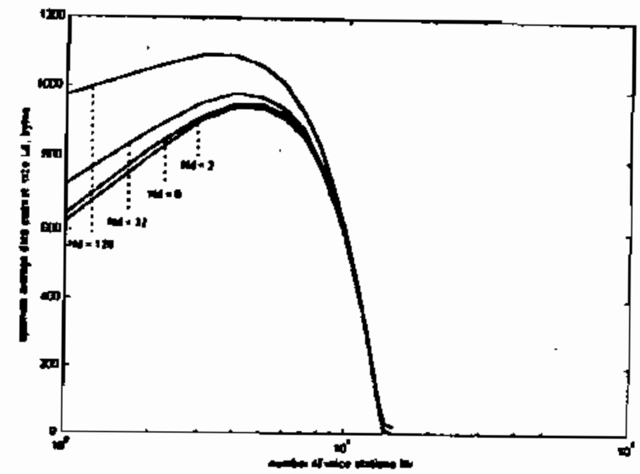

Fig.1 Optimum average data packet size as a function of the number of voice stations (C_r = 1 Mb/s)

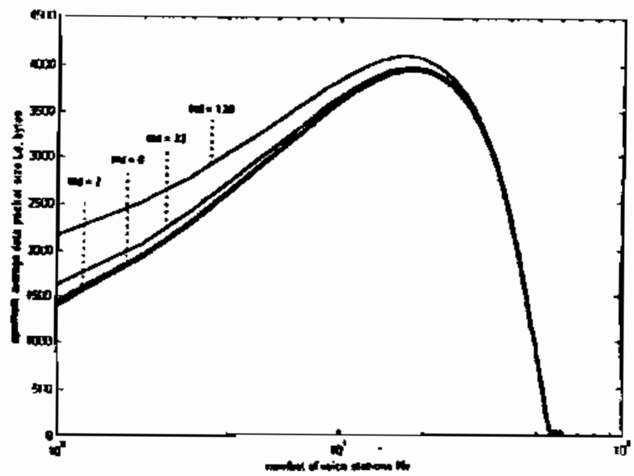

Fig.2 Optimum average data packet size as a function of the number of voice stations (C_r = 4 Mb/s)

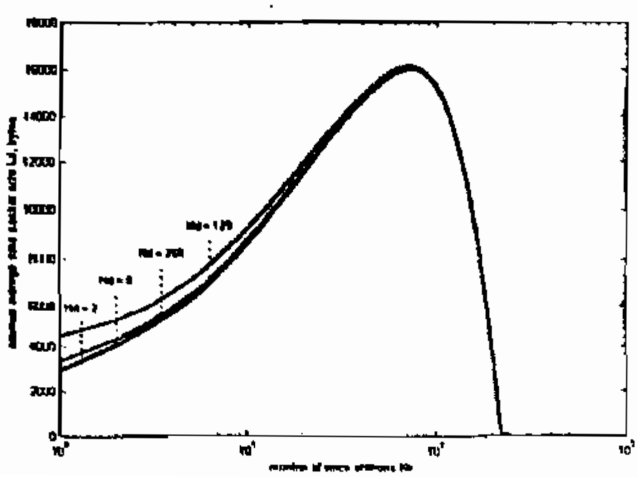

Fig.3 Optimum average data packet size as a function of the number of voice stations (C_r = 16 Mb/s)

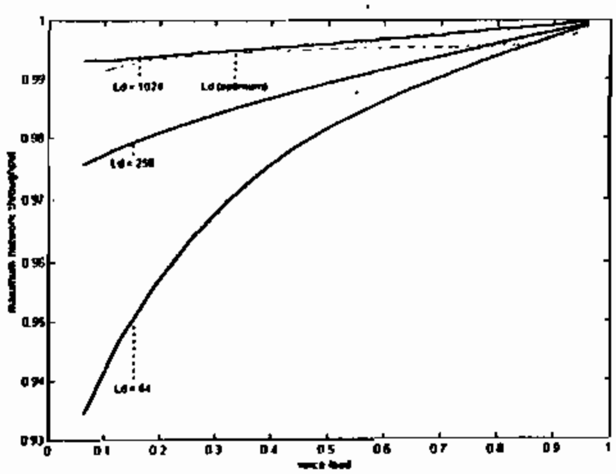

Fig.4 Maximum network throughput as a function of voice load (C_r = 1 Mb/s)

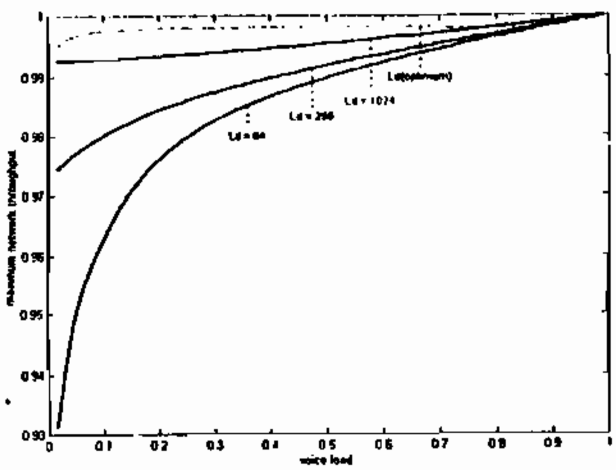

Fig.5 Maximum network throughput as a function of voice load (C_r = 4 Mb/s)

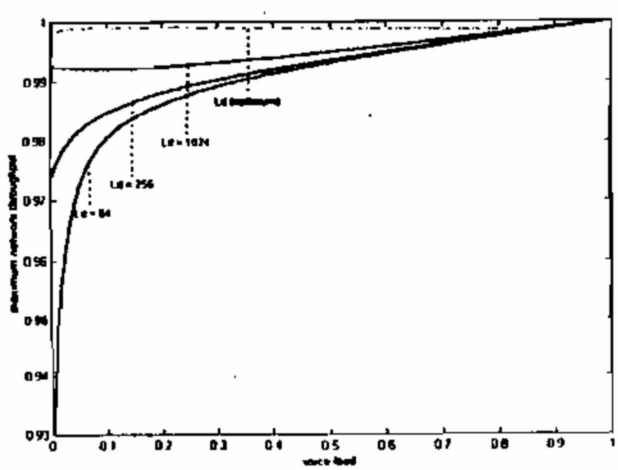

Fig.6 Maximum network throughput as a function of voice load (C_r = 16 Mb/s)

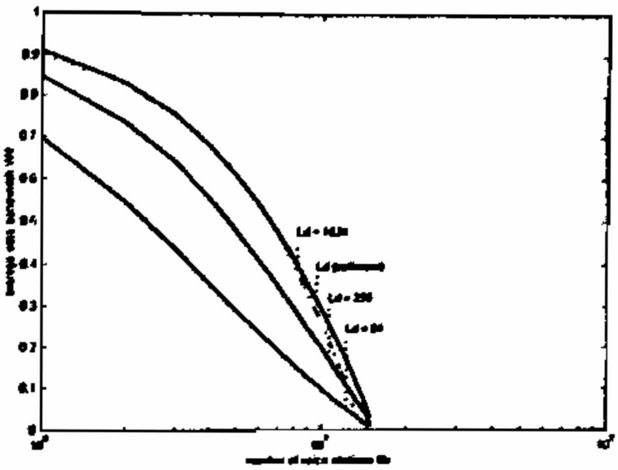

Fig.7 Average data bandwidth as a function of number of voice stations (C_r = 1 Mb/s)

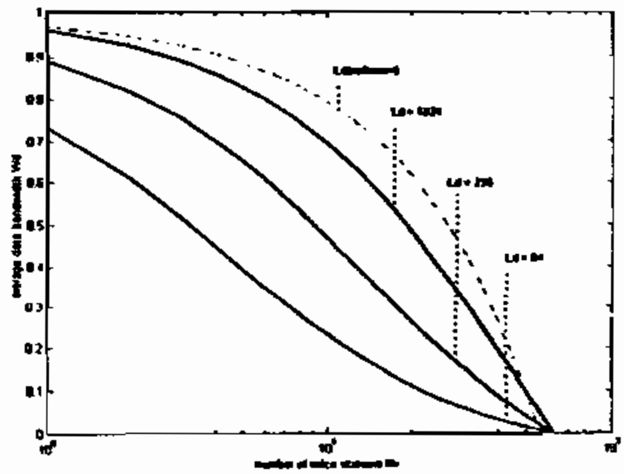

Fig.8 Average data bandwidth as a function of number of voice stations (C_r = 4 Mb/s)

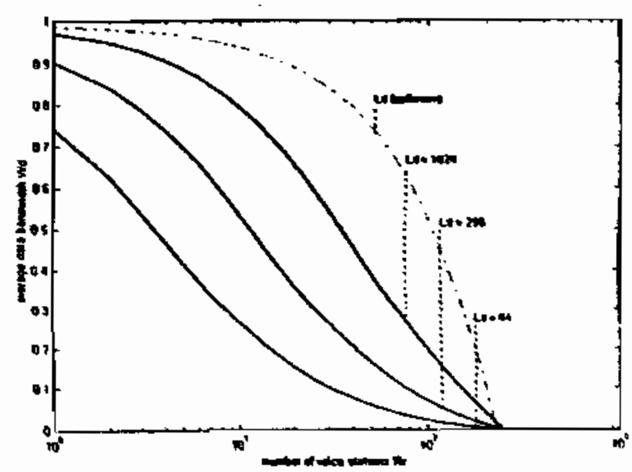

Fig.9 Average data bandwidth as a function of number of voice stations (C_r = 16 Mb/s)

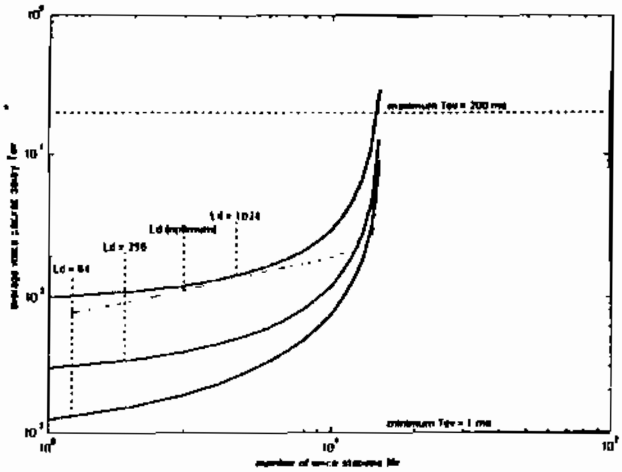

Fig.10 Average voice packet delay as a function of number of voice stations (C_r = 1 Mb/s)

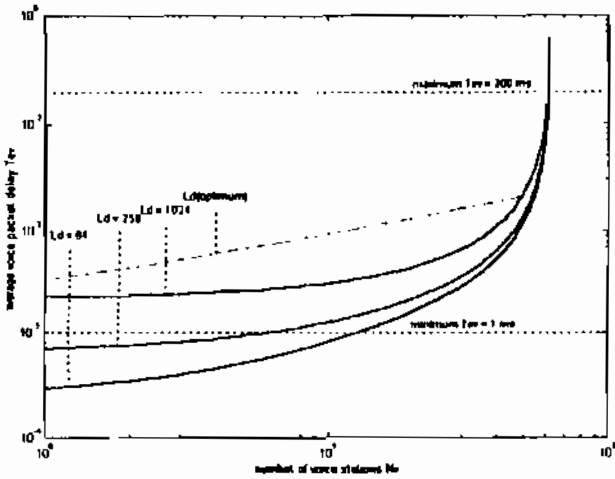

Fig.11 Average voice packet delay as a function of number of voice stations (C_r = 4 Mb/s)

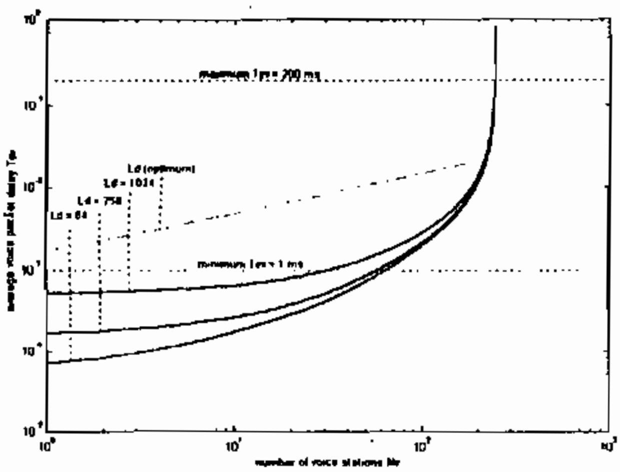

Fig.12 Average voice packet delay as a function of number of voice stations (C_r = 16 Mb/s)

VII. Conclusions

In this paper, simulation models were developed to optimize the performance of a token ring LAN for the integration of voice and data. As a result of this study the optimum average data packet size $L_{d(optimum)}$ is obtained. Moreover, it is shown that how the maximum number of voice users supported by the network is affected by the network bit rate.

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